3 Standard Methodologies

The objective of this section is to identify measurements and procedures performed repeatedly throughout the testing process, such as those for RF power measurement, audio S/N measurement and audio editing. The procedures for performing these operations are standardized for all tests, and are defined in the following subsections.

3.1 FM Band RF Measurements

This subsection defines procedures for measuring the RF power of FM Band signals.

3.1.1 FM Analog Power

Methodology

This procedure specifies the method for measuring analog FM power. An FM signal has the rather unique characteristic of constant power regardless of the content or amplitude of the modulating signal. Therefore, power can be measured with or without a modulating signal present.

If a modulating signal *is* present, then the power across the entire channel is integrated in order to determine overall power. Traditionally, there are three common laboratory methods of performing this integration:

1) Numerically, by utilizing a DSP based Vector Signal Analyzer (VSA) or band power markers on a spectrum analyzer
2) Physically, by detecting heat in a thermal sensor
3) Electrically, by using a diode to rectify the signal and take advantage of a diode's "square-law" region of operation

For our testing purposes, we shall use method (3) in conjunction with an average power meter and RMS responding diode detector.

Setup

The measuring instrument shall be an HP437B average power meter with an 8481D diode detection sensor, and will be configured as shown in Table 3-1.

Table 3-1 HP 437B Setup – Analog FM Power

Parameter	Description
Sensor Type	HP 8481D (diode detector)
Limit Checking	On
Low Limit	-70dBm
High Limit	-20dBm
Cal. Factor	98.5%
Note: "Preset" first, and then set above parameters	

Usage of the HP437B must take into consideration the dynamic range of the diode detector. Under no circumstances shall a measurement be taken outside the sensor's measurement range. In addition, extra care must be taken during IBOC measurements due to the high peak-to-average ratio of COFDM. Therefore, measurements with an IBOC signal must maintain 10dB of "headroom" below the sensor's peak range.

In addition, note that measurements must be made under conditions with no interferers present since any out of band signal will artificially increase readings on the power meter.

Procedure

1) Configure the instrument according to the tables found in the "Setup" section above.
2) The analog carrier can be either modulated or unmodulated. There is no procedural change for either case.
3) The power level shall be observed (or collected over the GPIB bus) and the instantaneous reading recorded.

Presentation of Data

The resulting measurement shall be expressed in dB units with a precision of 0.01dB

3.1.2 IBOC Hybrid Mode Power

Methodology

This procedure specifies the method for measuring the power of an IBOC signal in hybrid mode. A hybrid signal is the spectral sum of a traditional analog FM signal and COFDM digital carriers at the channel edges.

The true average power of this signal may be determined by integrating over the entire channel, using an average power meter or vector signal analyzer, as discussed in 3.1.1 above. However, the resulting number would not be easily related to traditional analog power measurements. For comparison purposes, we would like to be able to say that a –30dBm hybrid signal has the same amount of *analog* energy as a traditional –30dBm FM signal.

For this reason, all references to the power of a hybrid signal will actually be specifying power in the signal which results from traditional analog FM modulation. In other words, *a –30dBm hybrid signal will actually have –30dBm of analog FM energy plus the energy resulting from the digital carriers.*

Consequently, in order to exclude the digital energy from our measurements, it is not possible to directly measure hybrid mode power using a traditional average power meter (at least not without an impracticably steep bandpass filter). Therefore, there are three practical measurement methods that can be employed:

1) Remove digital sidebands and measure remaining analog power with a traditional average power meter or Vector Signal Analyzer
2) Use a Vector Signal Analyzer to numerically integrate over analog bandwidth
3) Use spectrum analyzer to measure power of the *unmodulated* analog carrier.

For our purposes, we will utilize the first method in conjunction with an HP437B average power meter. This is facilitated by the fact that the test bed has electromechanical RF switches, which can readily switch the digital carriers in and out of the spectrum. In addition, the removal of the digital sidebands increases measurement accuracy.

Setup

The measuring instrument shall be an HP437B average power meter with an RMS responding sensor, and will be configured as shown in Table 3-2.

Table 3-2 HP 437B Setup – Hybrid Mode Power

Parameter	Description
Sensor Type	HP 8481D (diode detector)
Limit Checking	On
Low Limit	-70dBm
High Limit	-20dBm
Cal. Factor	98.5%
Note: "Preset" first, and then set above parameters	

Usage of the HP437B must take into consideration the dynamic range of the diode detector. Under no circumstances shall a measurement be taken outside the sensor's measurement range. In addition, note that measurements must be made under conditions with no interferers present since any out of band signal will artificially increase readings on the power meter.

Procedure

1) Configure the instrument according to the tables found in the "Setup" section above.
2) The analog carrier can be either modulated or unmodulated. There is no procedural change for either case.
3) The digital carriers shall be removed by opening the appropriate electromechanical RF switch. The remaining energy will be purely analog FM.
4) The power level shall be observed (or collected over the GPIB bus) and the instantaneous reading recorded.

Presentation of Data

The resulting measurement shall be expressed in dBm, and rounded to the nearest tenth of a decimal place. It should be made clear that this power refers to the analog energy and does not represent the average power level of the entire hybrid signal (as discussed above).

3.1.3 Power in the Presence of Multipath

Methodology

This procedure defines the method which shall be used for determining the power of an analog or hybrid signal in the presence of multipath.

An FM signal undergoing dynamic multipath is inherently a signal whose power and spectral characteristics are time-variant. Any attempt to directly measure the power of such a signal would have to specify not only the power, but also the exact multipath conditions that existed at the moment of measurement. Since this is entirely impractical in a dynamic multipath simulation, an alternative approach must be devised.[3]

[3] For additional background information and discussion, please refer to ATTC Doc. 00-02, "The Measurement of Power as applied to IBOC DAB signals in the Presence of Multipath for the FM Band".

In order to facilitate such a measurement, all of the "echoes" or secondary paths are considered to be "Undesired" signals. The only "Desired" signal is the main path. Therefore, when a multipath impaired signal is said to have a signal strength of –47dBm, this implicitly means that the main path alone has a power of –47dBm.

Setup
The "bypass" multipath profile is used during this measurement. This consists of a single path with no loss, and is of the "Doppler" type with a 0Hz Doppler shift applied. Please refer to Appendix D for additional information on the bypass profile.

Procedure
1) Load the bypass profile into the multipath simulator
2) Start the simulation
3) Disable any AGC functionality, or set AGC to "hold".
4) For an analog signal, measure the power according to 3.1.1. For a hybrid signal, measure the power according to 3.1.2.
5) Load the multipath profile for the desired test (such as Urban Slow or Urban Fast), but make *no* additional adjustments (such as AGC settings) to the simulator.

Presentation of Data
The resulting power measurement shall comply to the format outlined in 3.1.1 or 3.1.2.

3.1.4 Ratio of Analog to Digital Power (IBOC Hybrid Mode)
Methodology
This procedure specifies the method used to determine the ratio of analog to digital power in a hybrid mode signal. In practice, the ratio of analog to digital power does not vary, and is not a user-defined parameter. It is a fixed parameter that is defined in the specifications and describes any hybrid mode signal. This measurement will therefore be performed periodically (during daily calibration) in order to verify that the test system is setup according to the definition of an iBiquity hybrid mode signal.

iBiquity Digital Corporation has specified that the average digital power shall be 20dB below the average analog power. It should be emphasized that this is the *average* power of the digital carriers, *not* the peak power.

The procedure is very similar to that discussed in 3.1.2. Electromechanical RF switches are used to separate the digital subcarriers from the analog energy. The digital energy is switched out, and the analog power is measured according to 3.1.1. Next, the digital energy is switched in and the analog switched out. The *average* digital energy is measured using an average power meter. The difference between these two measurements is verified to be 20dB.

Setup
The measuring instrument shall be an HP437B average power meter with an RMS responding sensor, and will be configured as shown in Table 3-3. The HP437B provides a higher level of accuracy than the VSA, and as such is the preferred instrument for analog-to-digital ratio measurements.

Table 3-3 HP 437B Setup – Analog-to-Digital Ratio

Parameter	Description
Sensor Type	HP 8481D (diode detector)
Limit Checking	On
Low Limit	-70dBm
High Limit	-20dBm
Cal. Factor	98.5%
Note: "Preset" first, and then set above parameters	

Usage of the HP437B must take into consideration the dynamic range of the diode detector. Under no circumstances shall a measurement be taken outside the sensor's measurement range. In addition, extra care must be taken during IBOC measurements due to the high peak-to-average ratio of COFDM. Therefore, measurements with an IBOC signal must maintain 10dB of "headroom" below the sensor's peak range.

Procedure

1) This measurement shall be made under conditions where no interferers are present since any out-of-band signal will artificially increase readings on the power meter.
2) The analog carrier can be either modulated or unmodulated. There is no procedural change for either case.
3) Remove the digital carriers by opening the appropriate electromechanical RF switch. The remaining energy will be purely analog FM.
4) Put the HP437B into "Relative" mode.
5) Remove the analog energy and re-insert the digital carriers via the electromechanical RF switches.
6) Record the "relative reading" on the HP437B. This is the ratio of analog-to-digital power.

Presentation of Data

The measurement resulting from step six (6) shall be expressed in dB units with a precision of 0.1dB.

3.1.5 Additive White Gaussian Noise Power

Methodology

This subsection describes the methods which shall be used to measure the average power of an Additive White Gaussian Noise (AWGN) signal.

There are several ways to express the power level of a noise signal such as Additive White Gaussian Noise (AWGN). The first method is bandwidth *independent*, and has units equal to degrees Kelvin. If an AWGN signal has a power that is said to be 30,000°K, then it has the same amount of noise power as a theoretical resistor would at a physical temperature of 30,000° Kelvin. The second method is bandwidth *dependent*, and has units equal to Power per Unit Bandwidth. For our purposes, we need to measure noise power in a given bandwidth. Therefore, we shall use the latter method, and express our results in dBm/Hz.

Unfortunately, the measurement of noise power in dBm/Hz units requires some rather specialized procedures and/or equipment. There are two well-known methods used to *measure* the power level of AWGN in bandwidth dependent units.

The first method is informally referred to as the "ENB Method". ENB is an abbreviation for *Equivalent Noise Bandwidth.* In reality, ENB is a property of a bandpass filter. Any bandpass filter has a passband with a finite bandwidth, which may be measured and characterized using network analyzers or similar instrumentation. This characterization may also involve the calculation of an ENB value for the filter (The details of ENB calculation are beyond the scope of this test procedure). This ENB calculation may then be used as a conversion factor to convert absolute power units (dBm) into power per unit bandwidth units (dBm/Hz). Using this methodology, one can band-limit AWGN noise using an ENB calibrated filter, and measure the resultant power in absolute units (using standard instrumentation such as a thermal power meter). These dBm units may then be converted to dBm/Hz units using the ENB conversion factor as follows:

$$\text{dBm/Hz} = \text{dBm} - 10\log(\text{ENB})$$

where ENB is the bandpass filter's Equivalent Noise Bandwidth, given in Hz units

The setup and procedures for performing a measurement using the ENB method are given in the sections below.

The second method of measuring noise power in dBm/Hz units employs an HP Vector Signal Analyzer (VSA). The VSA is a very sophisticated and rather unique device, because it is fundamentally a time domain device that performs rapid FFT's for spectrum analysis. This architecture allows one to easily measure Power Spectral Density (PSD) without the normal spectrum analyzer concerns of calibrated resolution bandwidth filters. Since the PSD measurement provides results in dBm/Hz units, noise power density can also be directly measured in dBm/Hz units. The setup and procedures for this measurement methodology are also given below.

Unfortunately, both of these measurement methodologies (ENB and VSA) are fundamentally limited by the low level sensitivity of the instrumentation (either the power meter or the VSA). For example, a 30,000K noise signal has a power level which is rapidly approaching the inherent noise floor of the VSA, and is also significantly below the limits of any thermal power sensor. In order to overcome this limitation, noise power is measured at a much higher power level. Then, in order to achieve lower noise power levels, a known/calibrated attenuator is inserted in-line with the high level noise source. (Insertion of this inline attenuation is facilitated by our noise generation instrument, which contains precision attenuators installed internally).

Setup (ENB Method)

The power measuring instrument shall be an HP437B average power meter with an 8481D diode detection sensor, and will be configured as shown in Table 3-4.

Table 3-4 HP 437B Setup – Analog FM Power

Parameter	Description
Sensor Type	HP 8481D (diode detector)

Parameter	Description
Limit Checking	On
Low Limit	-70dBm
Filter/Averaging	Auto
Note: "Preset" first, and then set above parameters	

Because the AWGN noise will have a relatively high peak-to-average ratio, the power sensor must operate with at least 10dB of "headroom". Therefore, this sensor may not be used to measure any AWGN noise power exceeding –30dBm.

Procedure (ENB Method)

1) Set the noise generator's output attenuator to 0.0dB, and enable the calibrated internal FM Bandpass Filter (FM BPF).
2) Remove all other bandpass filters from the noise signal path, and replace with a "barrel" connector (Refer to the test bed schematics for locations of additional filters).
3) Measure the absolute power of the noise, using the instrumentation described above, and observing the appropriate power limits.
4) Measure the insertion loss (at 97.9MHz) of any filters which may have been removed in Step 2. Subtract the insertion loss of these filters from the measurement of Step 3.
5) Convert the result of Step 4 from dBm to dBm/Hz units using the equation described in the methodology section. Note that the ENB of the noise generator's filter is available on the unit's calibration certificates.
6) Replace any bandpass filters which may have been removed in Step 2.

Presentation of Data (ENB Method)

The result of Step 5, above, shall be given in dBm/Hz units and expressed with a precision of 0.1dB.

Setup (VSA Method)

The power measuring instrument shall be an HP89441A Vector Signal Analyzer (VSA), and will be configured as shown in Table 3-5.

Table 3-5 HP Vector Signal Analyzer Setup – Noise Measurements

Parameter	Setting
Note: "Preset" first, and then set parameters below	
Center Frequency	97.9 MHz
Span	1.0 MHz
Range (sensitivity)	Highest sensitivity before "over" LED
Resolution Bandwidth	10kHz (auto-coupled)
Measurement Data	PSD (Power Spectral Density)
Averages	250
Marker Frequency	97.9MHz

Procedure (VSA Method)

1) Setup the HP 89441 VSA according to Table 3-5.
2) Note that the marker must be positioned to take measurements at 97.9MHz

3) Set the noise generator output attenuator to 0.0dB, and the FM bandpass filter inline/enabled.
4) Observe the power spectral density (PSD) marker reading on the VSA, ensuring that the running average has reached at least 250.

Presentation of Data (VSA Method)

The resulting measurement will be given in dBm/Hz units, and expressed with a precision of 0.1dB.

3.1.6 Deviation and Channel Configuration

Methodology

A typical broadcast FM signal has numerous configuration parameters, which need to be regularly monitored and verified. Some of these parameters include main channel modulation, pilot injection, SCA injection and SCA modulation.

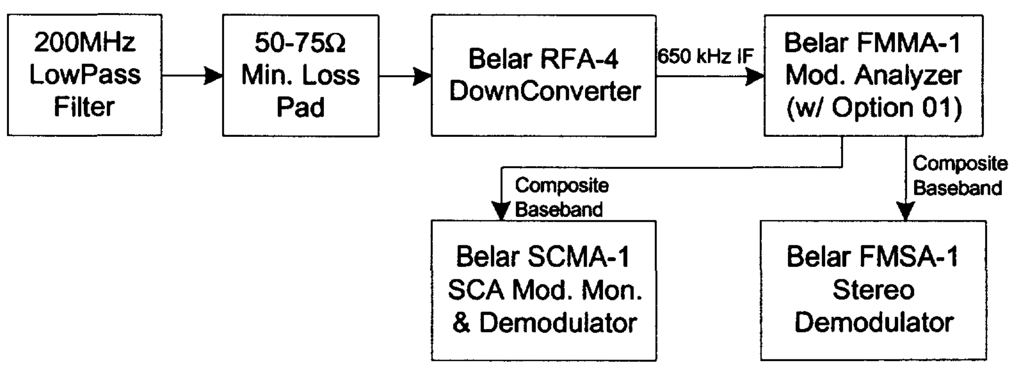
Throughout the testing process, there is a need to measure and verify these parameters. In recent years, the market has seen the introduction of broadcast grade modulation monitors that utilize DSP technology for more accurate measurements. These units reduce calibration drift and uncertainty. In addition, the digital readouts remove much of the guesswork from directly reading an analog deflection meter. For these reasons, the testing process shall utilize broadcast grade, DSP based modulation instruments to measure the various FM channel modulation parameters.

The performance of these modulation monitors shall be verified during the proof-of-performance calibration, using Bessel null methods where appropriate. For details on this verification methodology, please refer to the proof-of-performance plan, as referenced in section 1.1.2.

Setup

The FM channel characteristics shall be measured using the instrumentation and setup shown in Figure 3-1. A complete listing of the setup of these devices may be found in the test bed proof-of-performance record of test results, as referenced above in section 1.1.2.

Figure 3-1 Setup for Modulation & Configuration Readings



(Note: The 200MHz Low Pass Filter blocks the L.O. signal, which is otherwise reflected out of the RFA-4 input)

Procedure

The equipment shown in Figure 3-1 shall be used to make measurements as follows:

1) Belar FMMA-1
   a) Total peak modulation
2) Belar FMSA-1
   a) Pilot injection
   b) Individual left and right modulation
3) Belar SCMA-1
   a) Subcarrier injection
   b) Subcarrier percent modulation

It is important to note that these measurements may only be performed on analog signals. In other words, all measurements must be made with the *IBOC sidebands removed.* These instruments are not designed to provide accurate readings in the presence of IBOC sidebands, and erroneous readings will result if such a measurement is attempted.

Presentation of Data

The resulting data shall be expressed in percentage (%) units with a precision of 0.1%.

3.2 Baseband Audio Measurements

This subsection defines procedures to be used in measuring the characteristics of baseband analog audio.

3.2.1 Signal-to-Noise (S/N)

Methodology

This procedure defines the method for finding the S/N ratio of demodulated analog audio, as seen at the output of an FM receiver.

S/N is the ratio of some reference signal to the broadband noise floor of the Device Under Test (DUT). In our case, the DUT is not only the receiver, but also the entire broadcast system and communications channel (including interferers). For our purposes, the reference signal is defined as the *desired signal* in the interference scenario under test. Refer to the subsections of 2.1 for the definitions of various desired signals.

The noise measurements shall comply with the ITU-R 468.4 standard[4]. This standard defines the characteristics of a "quasi-peak" voltmeter to be used for detection, and also a filter that is used to weight the response. The weighting filter is an attempt to model the human auditory system. The human ear, for example, is not as sensitive to low frequency noise as it is to mid frequency noise. Therefore low frequencies carry less weight in a S/N measurement that adopts the 468 method.

In addition to 468 weighting, an additional 19kHz Low Pass Filter (LPF) will be used for measurements performed on the main channel of an FM receiver. The purpose of this filter

[4] International Telecommunications Union, ITU Rec.468-4, "Measurement of Audio-Frequency Noise Voltage Level in Sound Broadcasting"

is to remove any 19kHz pilot, which may remain at the output of the receiver. Since the pilot is at a frequency above the hearing limits of most individuals, it is normally not heard by the listener and does not significantly contribute to audible noise. However, it does degrade the accuracy of S/N measurements on the system. Therefore, the 19kHz LPF is used in order to increase the accuracy of these S/N measurements.

Setup

The measuring instrument shall be an Audio Precision System One, with the analyzer configured as shown in Table 3-6.

Table 3-6 Audio Precision Setup for S/N Measurements

Analyzer Settings		
Measure	A	Amplitude
Range	Auto	
BP/BR Freq.	Auto	
Detector	Q-Peak	
Bandwidth	10Hz	22kHz
Filter	CCIR-468	
Channel A	Input	100kΩ
Range	Auto	
Channel B	Input	100kΩ
Range	Auto	

Procedure

1) Apply the desired signal to the receiver. This will be the *reference signal* for the S/N measurement. (Note: ensure that all audio processing is disabled by setting the FM processor to "bypass" mode)
2) Set up the Audio Precision analyzer according to Table 3-6.
3) Use the Audio Precision analyzer to measure the value of the voltage generated across the appropriate output of the receiver. Record this value for use in step 7.
4) Remove modulating audio from the input to the stereo generator. Note that the stereo pilot and subcarriers remain – only the audio input is removed from the stereo generator.
5) Use the Audio Precision analyzer to measure the value of the voltage remaining at the output of the receiver. Repeat this measurement until 40 readings are obtained.
6) Find the statistical mean of the 40 readings.
7) Find the ratio of the result from step 3 (in volts) to the result from step 6 (also in volts).

Presentation of Data

The ratio resulting from step 7 (above) is a unit-less quantity, which should be expressed in dB. Since this ratio is derived from voltage units, the calculation shall be: *20*log(ratio).* The result shall be rounded to the nearest tenth of a dB.

3.3 Audio Recording and Editing

This subsection defines procedures for performing common audio editing operations, such as digital recording, editing and leveling.

3.3.1 Audio Recording

Methodology

All audio recordings must be made in such a way that no significant artifacts are introduced by the recording process. This necessitates the exclusive use of a digital audio recording format. Furthermore, this format must be uncompressed and able to sustain a data rate that supports multi-track audio with a resolution of 16 bits and a sampling frequency of 44.1kHz. Additionally, the recordings must be made in a manner that lends itself to archival and duplication.

In order to meet these requirements, the DTRS digital tape recording technology of Tascam shall be used. This format records eight tracks of digital audio on the same cassette shell utilized by the popular Hi-8 video format. Tascam's format has also gained widespread popularity within the professional audio recording community, enabling easy exchange of materials across different facilities.

There is also a requirement to unambiguously associate individual "takes" on any given tape with a specific test number or test setup. In order to accomplish this objective, SMPTE timecode shall be employed extensively. The DA-98 recorder has provisions for a ninth track, containing unique SMPTE timecode. This timecode provides the ability to log the contents of a DTRS digital audio tape with resolution to 33.4 milliseconds.

In order to generate this log of tape contents, identical and synchronous time code shall be routed to the DA-98 recorder and an external computer simultaneously. The external computer shall keep track of the current test setup and take a snapshot of the SMPTE timecode at the start of each test. Microsoft Excel Visual Basic scripts shall then be used to generate a log, which will relate SMPTE timecode locations to specific test conditions.

Setup

The recording device shall be a Tascam DA-98 Digital Multitrack Recorder, operating in conjunction with a Tascam IF-AE8 AES digital audio interface. Complete details of the menu setups for these units may be found in the test bed proof-of-performance documentation.

Procedure

Operation of the Tascam DA-98 recorder is straightforward. However, there are several important points which shall be observed by the test engineer throughout the testing process:

1) The recorder shall always be operated with a 44.1kHz sampling frequency and 16 bits of resolution
2) Whenever possible, the digital inputs of the IF-AE8 AES interface shall be used. Normally, sample rate conversion shall be disabled. However, certain sources containing a high level of jitter may require activation of sample rate conversion.
3) For sources available only in an analog format (e.g. radio outputs), the balanced analog inputs of the DA-98 shall be employed, utilizing the unit's internal A-to-D converters.

Presentation of Data

1) *All* tapes shall be labelled with a unique identification code consisting of the date and an incrementing index number (e.g.: 07-25-01-03 represents the third tape generated on 07-25-01).
2) *All original* test results tapes shall be archived in ATTC's tape vault, and indexed in ATTC's tape library database.
3) A log shall be generated for each recording. This log shall relate SMPTE timecode values and tape identification numbers to test setup conditions.

3.3.2 Audio Editing

Methodology

The "raw" test result tapes generated by the test bed are expected to require significant editing. This editing should eliminate periods of silence and any glitches that may occur while equipment setups are being changed in between tests. The editing should occur in a manner which introduces *no* additional audio artifacts or impairments. Additionally, the editing should preserve the original tape in its entirety. The recording from each test shall be formatted into an individual computer file, which can then be readily played back on the equipment of the subjective evaluation laboratory.

In order to accomplish these objectives, it is necessary to perform the editing in the digital domain on a digital audio workstation. This workstation must provide facilities for transferring DTRS tapes to computer format .wav files. It must also provide professional audio editing software, and a CD data recorder to export the resultant .wav format files.

Setup

A digital audio workstation shall be used for all editing. This workstation shall consist of:

1) High speed PC workstation
2) Lynx One professional audio card w/ AES and word clock I/O
3) Lucid DA9624 external D/A converter and headphone amplifier
4) Sennheiser HD-600 headphones
5) Horita TR-100 stand-alone time code reader
6) Tascam TDIF interface card
7) Tascam DA-98 or DA-78HR DTRS player/recorder
8) Cool Edit Pro professional audio editing software

Procedure

1) Transfer DTRS digital audio tape to the computer hard drive and save in .wav format (for details, refer to: *DTRS Tape to .wav Transfer Procedures, ATTC Doc. 01-18*)
2) Use SMPTE time code to locate each test within the original .wav file
3) Trim/edit the "heads" and "tails" of each test, such that there is absolutely no silence before or after each test.
4) Save this trimmed file, using a filename that conforms to the following format: *ATTCTest#_RadioAbbreviation_AudioCut.wav*

Presentation of Data
All edited audio files shall be transferred to a recordable CD in .wav format. These audio files shall be archived at ATTC with a log relating the .wav filename, the ATTC test number and the actual test setup conditions.

3.3.3 Audio Leveling & File Renaming
Methodology
Once the test results audio has been recorded, transferred to .wav format and edited, there is an additional requirement to "level" this audio for certain applications. The subjective evaluation laboratory has designed experiments that require all audio samples to have approximately the same perceptual loudness. For example, if an experiment participant is required to listen to a group of audio samples, each one of these samples should have the same perceptual loudness as the other samples in the group.

In order to meet this objective, an additional editing step is required. This additional step is referred to as "leveling". The leveling process utilizes audio editing software to adjust the amplitude of audio recordings. A professional audio editor subjectively evaluates the perceived loudness of each sample within a group, and then adjusts all samples to fall within an acceptable range of perceptual loudness.

In addition to this leveling process, the files must also be renamed in such a way that their filename represents the most important parts of the test setup. Although the ATTC test number uniquely and fully identifies each test, a more descriptive, alphanumeric filename is useful for the subsequent data analysis stages. A convention for this descriptive filename is described in the document entitled: *Naming Convention for Subjective Audio Files, ATTC Doc. 01-03.* A Visual Basic computer program shall be used to convert the original filename (e.g. *5102_Delp_Prince.wav*) into the more descriptive filename (e.g. *5102_B_Delp_NONE_X_NONE_X_AWGN_B-2dB_X_PRINCE_ROCK.wav*). This Visual Basic program shall employ a lookup table to relate the ATTC test number in the original filename to the new descriptive filename, and then perform the rename operation.

Setup
The equipment setup shall be identical to the editing setup described in 3.3.2. It is important to note that the external D/A converter and headphones are identical to the equipment used in the listening stations of the subjective evaluation laboratory. In this manner, the professional audio editor evaluates perceptual loudness using an equipment setup that emulates the setup used by experiment participants.

Procedure
1) Determine which audio samples will belong to the experimental group. (This information will be provided by the experiment designer)
2) Copy all of these audio samples into a common directory. (Note that these samples should all have been recorded, transferred and edited in accordance with the procedures of 3.3.1 and 3.3.2.)
3) Use the Windows Commander software application to listen to the first few seconds of each audio sample in the experiment.

4) Make notes on the perceptual loudness of each sample, and estimate the relative loudness of this sample compared to the quietest samples in the group. These estimations should be in dB units of attenuation required.
5) Review estimated attenuation values for any discernible pattern. (This pattern may relate to the audio cut, the radio type, the interference conditions, etc...)
6) Adjust each audio sample according to these estimates or any pattern that may have been identified.
7) Repeat steps three thru six until all audio samples in the group have nearly the same perceptual loudness. (Note that the audio editor should constantly strive to minimize the number of amplitude changes that are made to each audio sample, as excessive amplitude adjustments may add a minute amount of noise to the signal.)
8) Use the appropriate Visual Basic script and lookup table to automatically rename the leveled audio files with their more descriptive filename.

Presentation of Data

All leveled audio files shall be transferred to a recordable CD in .wav format. These audio files shall be archived at ATTC, and copies may be sent to the subjective evaluation laboratory. Note that the filenames shall contain the original ATTC test number, so the logs generated from the editing procedure of 3.3.2 may also be used to identify these leveled audio cuts.